**CLAIM AMENDMENTS**

IN THE CLAIMS

This listing of the claims will replace all prior versions, and listing, of claims in the application or previous response to office action:

1. (Previously Presented) A method for testing the transmission quality of a bidirectional real speech transmission or multicast connection over an IP network between a first VolP endpoint and a second VolP endpoint, comprising:

transmitting RTP speech packets from the first to the second VolP endpoints, and transmitting RTP speech packets from the second to the first VolP endpoints;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number; and

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality.

2. (Previously Presented) The method as claimed in claim 1, wherein the predetermined time period for a 10 Mbit/s transmission channel is longer that 5 s.

3-4. (Cancelled)

5. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is subjected to a threshold value discrimination to suppress side effects due to features of a communication protocol.

6. (Previously Presented) The method as claimed in claim 22, wherein quotients outside a predetermined tolerance range around the value 1 are valid as a representation of a reduced transmission quality.

7. (Previously Presented) The method as claimed in claim 23, wherein difference values outside a predetermined tolerance range around the value 0 are valid as a representation of a reduced transmission quality.

8. (Previously Presented) The method as claimed in claim 1, wherein the detected first and second numbers and/or the calculated values for a plurality of first and second VoIP endpoints connected to the IP network between which bidirectional speech connections exist in each case are logged.

9. (Previously Presented) The method as claimed in claim 1, wherein the detected first and second numbers and/or the calculated values for a plurality of first and second VoIP endpoints connected to the IP network within which bidirectional speech connections exist in each case are subjected to summarizing statistical processing to obtain an overall value representing the overall transmission quality of the IP network or of a section of the overall transmission quality of the IP Network.

10. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is signaled to subscribers at the first and/or second VoIP endpoints and/or to an operation control center of the IP network.

11. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is used as an input variable for controlling the speech transmission over the IP network.

12. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is determined substantially in real time and is signaled or is used as an input variable for controlling the speech transmission.

13. (Previously Presented) The method according to claim 2, wherein the predetermined time period is in the range of about 10 s to 30 s.

14. **(Currently Amended)** A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting RTP speech packets from the first to the second VoIP endpoints, and transmitting RTP speech packets from the second to the first VoIP endpoints;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number;

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality; and

routing **[[the]]**__**a**__ connection between the first and second VoIP endpoints based on the value.

15. (Previously Presented) A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting RTP speech packets from the first to the second VoIP endpoints,

transmitting RTP speech packets from the second to the first VoIP endpoint;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number;

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality; and

setting transmission parameters based on the value.

16. (Previously Presented) A system, comprising:

a detecting unit, arranged at a detection point on a transmission channel between a first and a second VoIP endpoints, to detect an enumeration of RTP speech packets transmitted from the first to the second VoIP endpoints as a first number, and to detect an enumeration of RTP speech packets transmitted from the second to the first VoIP endpoints as a second number;

an arithmetic processing unit connected on the input side to the detecting unit to calculate a value representing the transmission quality from the first and second numbers.

17. (Previously Presented) The system as claimed in claim 16, wherein the arithmetic processing unit has a division or subtraction stage.

18. (Previously Presented) The system as claimed in claim 16, wherein connected downstream of the arithmetic processing unit is a threshold value discriminator to evaluate the value representing the transmission quality with the aid of at least one predetermined threshold value.

19. (Previously Presented) The system as claimed in claim 16, further comprising a storage device connected on the input side to the output of the detecting device and/or of the arithmetic processing unit to log the first and second numbers and/or the calculated values.

20. (Previously Presented) The system as claimed in claim 16, further comprising a statistical processing unit, connected on the input side to the output of the detecting device and/or of the arithmetic processing unit, to summarize statistical processing of the detected numbers or calculated values in order to evaluate the overall transmission quality of the IP network or of a section of the same.

21. (Previously Presented) The system as claimed in claim 16, further comprising a signaling device to signal the calculated value or the overall value to the subscribers at the first and/or second VoIP endpoint and/or to an operation control center of the IP network.

22. (Previously Presented) The method as claimed in claim 1, wherein the arithmetic processing includes a division, where a value 1 of the quotient represents the highest transmission quality.

23. (Previously Presented) The method as claimed in claim 1, wherein the arithmetic processing includes a subtraction, where a value 0 for the difference represents the highest transmission quality.